Basic Principles of Stereophonic Sound

By WILLIAM B. SNOW

Stereophonic sound has become of vital importance to industry. The subject has been studied for many years, but the published material is scattered. This paper summarizes the fundamental theory underlying stereophonic sound so far as it has been published, and gives examples of how the theory is employed in representative practical situations. Fundamental differences between ordinary binaural listening and stereophony are pointed out, as well as similarities. It is shown that much qualitative but little quantitative information has been reported. Factors which aid some stereophonic effects are shown to be detrimental to others, and methods of minimizing the undesirable conditions are suggested. Applications to recording are discussed.

In 1941 K. de Boer wrote: "When the time comes to make use of stereophonic reproduction in the cinema, in broadcasting, etc., and the opinion becomes more and more general that the improvement in quality so obtained is worth the trouble, it will become necessary in the first place to find a process of making stereophonic records on a large scale."[6] Although even at that time stereophonic reproduction was far from new,[19–21] de Boer's enthusiasm for "making an orchestra plastically audible"[5] was shared by only a few. Now the time he forecast has finally come. Stereophonic sound has suddenly become of vital concern to the motion-picture and sound-recording industries, with multiple-channel recording the order of the day. This great upsurge of interest encouraged the preparation of this review of basic principles, and bibliography, as a guide for the large number of engineers who must quickly put this new technique into everyday use.

Stereophonic reproduction brings a truly remarkable increase in the realism of the sound and in the pleasure of listening to it. In one attempt to measure this quantitatively, reported by Fletcher,[2] the observers listened alternately to single-channel and stereophonic reproduction. In the stereophonic channels low-pass filters were inserted, while the single channel was maintained flat

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to 15 kc. Half of the observers still preferred stereophonic reproduction when the low-pass cutoff was reduced to about 5 kc. However, this paper is concerned primarily with the mechanism of stereophonic sound rather than its advantages, which are now so well recognized. It is not the purpose here to repeat detailed discussions that can readily be found in the references. Such data are summarized, and additional interpretation is provided. The serious reader is strongly urged to study the references carefully; a good grounding in this complicated subject can be obtained only in this way.

DEFINITIONS

As in most new developments, differences in nomenclature have arisen which tend to obscure precise descriptions of systems. The words "binaural" and "stereophonic" are those most frequently used, but not with uniform meanings. This is not a new phenomenon. Alexander Graham Bell, writing in 1880,[1] referred to the "stereophonic phenomena of binaural audition," in describing experiments on the directional sense in hearing conducted with his newly invented telephone. The following definitions apply to the discussions of this paper and are limited to electro-acoustic sound-reproducing systems:

Binaural — A system employing two microphones, preferably in an artificial head, two independent amplifying channels, and two independent headphones for each observer. This duplicates normal listening.

Stereophonic — A system employing two or more microphones spaced in front of a pickup area, connected by independent amplifying channels to two or more loudspeakers spaced in front of a listening area. This creates the illusion of sounds having direction and depth in the area between the loudspeakers.

It is very important to distinguish between these systems. A binaural transmission system actually *duplicates* in the listener's ears the sounds he would hear at the pickup point, and except that he cannot turn the dummy head, gives full normal directional sense in all directions. A stereophonic system produces an abnormal sound pattern at the listener's ears which his hearing sense *interprets* as indicating direction in the limited space between the loudspeakers. It has been aptly said that the binaural system transports the listener to the original scene, whereas the stereophonic system transports the sound source to the listener's room.

ELECTRO-ACOUSTIC SOUND-REPRODUCING SYSTEMS

Outstanding differences and similarities of the various types of electro-acoustic reproducing systems are summarized in the chart of Fig. 1. The "System" names in column 1 conform to a uniform pattern and will be found in the literature, except "Monophonic" which is used for convenience as the opposite of stereophonic. "Equivalent Normal Experience" refers to the everyday hearing experience that most closely parallels listening over the systems in question. The next four columns are obvious. The column "Direct Sound Reproduction of Single Source Pulse" is probably the most important, since it gives the basic differences between the sound produced by the various systems. If a single sound pulse is produced by the source, this column gives the characteristics of the resulting *direct* sound pulses at the observer's ears. The direct sound is the initial sound transmitted directly from source to observer by the shortest path, and arriving before any reflected sound arrives. It has been found that the direct sound carries the information, making angular perception possible, and it will be referred to frequently in what follows. Reverberant sound arrives from many angles and confuses the directional perception if too great in intensity. The "Remarks" column gives qualifying comments concerning the sound reproduction of each system. The reasoning behind these remarks is given in the body of the paper.

BINAURAL REPRODUCTION

Binaural reproduction as used herein means ordinary two-ear listening since the reproducing system transmits a faithful copy of the original sound to the listener's ears.

Angular Localization

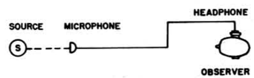
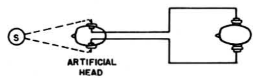
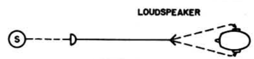
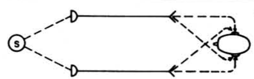
The properties of hearing which give the directional sense in binaural listening have been studied extensively.[11-18] For pure tones, angular localization is produced partially by phase differences at the two ears caused by the difference in distance from source to the ears as the source angle changes. The phase effect becomes ambiguous somewhat above 1000 cycles because at short wavelengths more than one angle results in the same phase difference. However, in the higher-frequency region intensity differences produced by the diffraction or sound-shadow effects of the head and external ears become great enough to give angular localization.

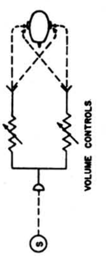
The great majority of sounds are not pure tones, but complex. For complex sounds the equivalent effects are arrival time and quality difference. A complex wave pulse has an initial wavefront which arrives at the near ear a short time before it arrives at the far ear. It is this small time difference which is used by the hearing sense to determine small angular variations, particularly for sounds near the median plane (straight ahead). It is characteristic to turn toward a source to locate it with maximum precision, and for impulsive sounds such as speech or clicks, differences as small as 1° to 2° can be perceived. These angles correspond to arrival-time differences of about 10 to 20 μsec, and the maximum possible difference, for a source in line with the two ears, is only about 700 μsec. The loudness differences at such small angles are negligible and it must be assumed that the arrival-time differences give the localization clues. On the other hand, it is not possible for the mechanism of a *single* ear to distinguish such short time intervals[17]; this "decoding" of the arrival time differences must be accomplished by the brain.

The arrival-time effect is aided by the quality differences at the ears caused by sound diffraction.[22] Quality difference is another way of saying that a change in waveshape is produced. The intensity differences due to diffraction are functions of frequency and cause a complex sound to have a different frequency-intensity composition or quality at each ear. It is undoubtedly this effect which removes ambiguities in direction which would result from arrival time alone, because the diffraction effects are so complicated that a given quality difference can correspond only to one direction. Quality differences also change most rapidly near the median direction; consequently, angular localization is much less precise at the side than in front or back.

Changes in both arrival time and quality are relatively small as a source is elevated in front of an observer. Therefore the ability to distinguish angle in the vertical direction is relatively poor.

Fig. 1. Electro-acoustic sound reproducing systems.

Type of system	Equivalent normal experience	Pickup	Number of channels	Reproducer	Symbolic schematic	Direct sound reproduction of single source pulse	Remarks
Monaural	One ear plugged	1 Microphone	1	1 Headphone		1 Pulse to 1 ear	Only partial equivalence, as directional effects are not duplicated
Diotic	Sound source in median plane only	1 Microphone	1	2 Headphones		1 Pulse to each ear 0 time difference No quality difference	Only partial, as directional effects of elevation not duplicated. No directional effect of room reverberation Qualitatively much superior to monaural
Binaural	Normal listening, head stationary	2 Microphones in artificial head	2	2 Headphones		1 Pulse to each ear 0 to 0.6 msec time difference Normal quality differences	Directional and quality effects of normal listening in *pickup room duplicated* throughout *360° sphere* *No* listening room effects Observer cannot turn to face virtual source
"Monophonic" ordinary loudspeaker	Sound coming through a hole in a wall	1 Microphone	1	1 Loudspeaker		1 Pulse to each ear 0 time differences No quality differences	Both ears used—binaural listening to a single fixed sound source position Composite of pickup and listening room acoustics Listening room reverberation has directivity Represents widely spaced "effects" loudspeakers
Stereophonic	None	2 or more microphones	2 or more	2 or more loudspeakers		2 or more pulses to *each* ear 0 to 30+msec time differences Complex quality differences	Depends on fusion of multiple pulses by ears Directional and quality effects of normal listening *simulated* for a *limited angular area* appearing as part of *listening room* 360° simulation would require a large number of channels Composite of pickup and listening room acoustics Reverberation has partial directivity in pickup room, full directivity in listening room Observer can turn to face virtual source Time differences from both source and observer positions

Pseudo or bridged stereophonic	None	1 Microphone	2 or more	2 or more loudspeakers		2 or more pulses to *each* ear 0 to 20+ msec time differences Small quality differences	Single source only Many properties identical with true stereophonic system Time differences *only* from observer position, and detrimental Identical signal-reverberation pattern from each loudspeaker Identical quality from each loudspeaker Many other bridged combinations possible

The statement is made in Fig. 1 that the observer cannot turn to face the source. While systems have been constructed with servo connections between observer and dummy,[7] thereby improving localization, this is not practicable for a system used with multiple observers, or with a recording link.

Depth Localization

Perceiving the position of a sound source in space involves the determination of distance as well as angle. The ear has no mechanism corresponding to that of the eye for converging on the source, and must depend on less definite clues. In the absence of reverberation, the only information given is intensity and quality. From past experience the ear can form an approximate idea of distance from its interpretation of the *absolute* loudness of a sound, and from its judgment of quality differences produced by atmospheric absorption. These comparisons are made with a mental image of what the sound should be. In the presence of reverberation,[23] the ear can judge distance based on the ratio of direct to reverberant sound. Since neither of these methods is precise, judgment of distance is much less accurate than perception of angle. Probably everyone has had the experience of badly misjudging the distance of a sound heard for the first time, whereas no difficulty was experienced in determining its direction.

Fundamental Difference from Stereophonic Sound

This discussion of the determining physical factors underlying ordinary binaural hearing has been given at some length to lay a foundation for the discussion of those underlying stereophonic reproduction. There are basic differences which have been almost universally overlooked. When this confusion is cleared up, stereophonic reproduction can be used with much greater ease and satisfaction.

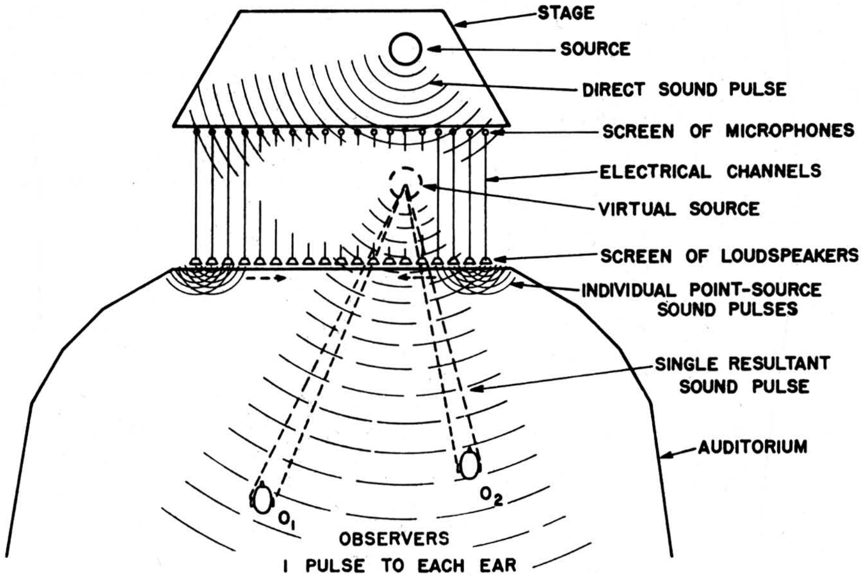


Fig. 2. Ideal stereophonic system. A very large number of very small microphones and loudspeakers would give a perfect reproduction of the original sound.

STEREOPHONIC REPRODUCTION

Fundamental Process

Publications. Good summaries of stereophonic sound are given by Frayne and Wolfe[24] and Knudsen and Harris.[25] Only a few reports on the fundamental principles of stereophonic reproduction have appeared in the literature[4,8,9,17,27,39] and these do not discuss identical operating systems. The Bell System tests and those at Twentieth Century-Fox Studios were made with widely spaced microphones, whereas scientists of the Philips Company employed closely spaced microphones, usually in an artificial head. It is unfortunate that additional fundamental tests made at Bell Telephone Laboratories were never reported in technical journals because of the press of other work and the advent of the War. In spite of this, we believe it is possible to understand the principles qualitatively, if not fully on a quantitative basis, and that the results so far published are for the most part consistent.

Screen Analogy. It has become customary to describe stereophonic reproduction as follows: A screen consisting of an extremely large number of extremely small microphones is hung in front of the sound source. Each microphone is connected to a corresponding extremely small loudspeaker in a screen of loudspeakers hung before the audience. Then the sound projected at the audience will be a faithful copy of the original sound and an observer will hear the sound in true auditory perspective. It is then stated that such an impractically large number of channels is not needed and that good auditory perspective can be achieved with only two or three channels. These are true statements, and the natural inference from their juxtaposition is that far less than faithful "space" reproduction of sound will give localization by ordinary binaural mechanisms. When we proposed this theory early in

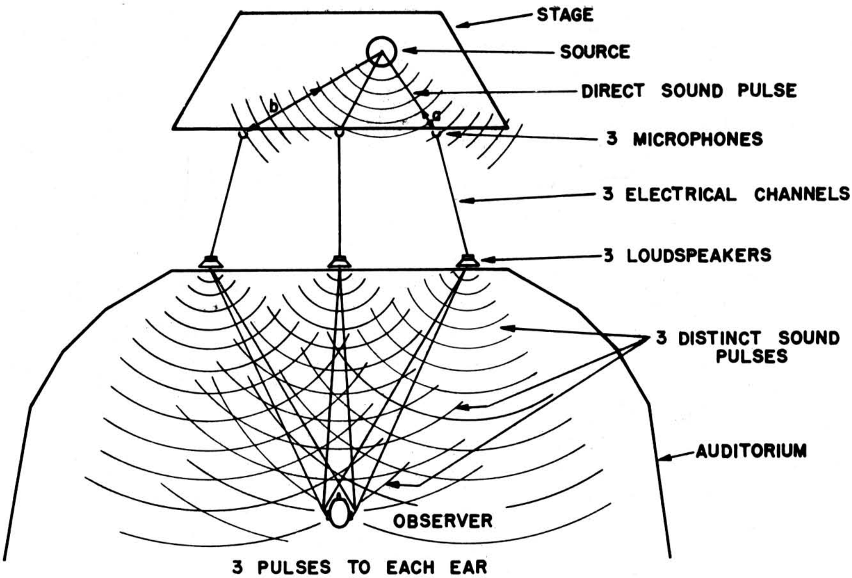


Fig. 3. Actual 3-channel stereophonic system. A practical stereophonic system gives a multiple reproduction of the original sound which the observer interprets as coming from a single source.

our studies of stereophonic phenomena, we realized that there were fundamental differences which were not fully understood, and pointed out the multiple source effect in connection with our loudness calculations.[26,27] Apparently this has not been sufficiently emphasized. The experience of the intervening twenty years has convinced this writer that this natural inference is mistaken, and has caused the confusion postulated in the previous section.

The myriad loudspeakers of the screen, acting as point sources of sound identical with the sounds heard by the microphones, would project a true copy of the original sound into the listening area. The observer *would* then employ ordinary binaural listening, and his ears would be stimulated by sounds *identical* to those he would have heard coming from the original sound source. As shown in Fig. 1, this means one direct-sound pulse to each ear for a single pulse from the source. The phenomena are illustrated schematically in Fig. 2.

Operating Conditions — Illusion Created. Figure 3 illustrates the conditions for a typical setup where only three channels are used. This arrangement does indeed give good auditory perspective, but what has not been generally appreciated is that conditions are now so different from the impractical "infinite screen" setup that a different hearing mechanism is used by the brain. Each individual loudspeaker sends a pulse to the observer. He therefore receives *three* faithful copies of the sound at *each* ear in rapid succession. The time differences between these pulses are too short to allow the ear to distinguish them as separate; consequently the hearing mechanism fuses them[17] into an *illusion* of a single sound pulse coming from a virtual sound source located somewhere in the space between the outer loudspeakers. The time differences are short, but still long compared to the maximum of 700 μsec to which the ears are accustomed in normal listening. Thus this type of listening falls outside of normal experience, but fortunately the brain is able to form a single concise impression from what might be expected to be a confusing set of signals sent by the ears.

The closest parallel is reverberation. But while there are distinct similarities, the three direct-sound pulses arrive ahead of any reflections other than the floor reflections which do not have individual directivity. In addition, they are separate and distinct, of high fidelity, and in a compact directive pattern. The reverberation follows as a "smear" of echoes of random directivity, and does not create a virtual source illusion.

The problem, then, in stereophonic reproduction is to produce multiple sound images at the ears of the observer which will fuse in such a way as to give the desired *illusion* of sound origin.

Angular Perception

Intensity Differences. What are the characteristics of the direct-sound pulses which cause them to give the observer the sensation of angular localization of the virtual source? The most obvious difference is intensity of sound projected by the several loudspeakers. These differences are caused by the varying distances of the source from the various microphones. When the source moves close to a microphone the output of the corresponding loudspeaker is greater than that of the other loudspeakers, and localization tends in its direction. The virtual source therefore moves in the same direction as the real source, and with proper system design can be made to have essentially proportional movement. In the original paper[27] Dr. Steinberg and this writer discussed this in detail and proposed a theory for the effect of these intensity differences, based upon the total loudness that would be produced in each ear by the total direct sound from all loudspeakers, taking into account the directivity of hearing caused by the shape of the head. While the agreement between the theory and experimental results was by no means perfect and the differences were pointed out, the theory did appear to account for the main effect. This theory has been questioned by other experimenters, principally, it is believed, because of the common confusion between the mechanisms of ordinary binaural hearing and stereophonic hearing which the discussion above should have now dispelled.

While a true understanding of the process is highly desirable, for the purposes of this paper it is not necessary to be certain of the precise physiological and psychological mechanisms involved. It is well established that intensity differences in the channels are an extremely important contributor to angular perception. With positions of source and observer fixed so that all other factors are constant, variation of the gain controls in the channels can shift the virtual source to any angular position in the reproducing area. This is true for any combination of source and observer positions. In practice this is important because gain is easily controlled, to correct faults in pickup, or to enhance angular movement. The bridged-microphone system of Fig. 1 operates on this basis, since the only differences that can be given the loudspeaker outputs must be obtained from electrical controls in the channels. As this is written, many pictures are being made "stereophonic" by the use of volume controls in bridged channels from sound tracks originally recorded for single channel or "monophonic" reproduction. The pseudo-stereophonic system has its place; but it is not a satisfactory substitute for a real stereophonic pickup. It does not have the benefit of the other aids to angular or depth perception described below; and in particular it can be used on only a single source at one time, so that an individual source and "pan-pot" must be supplied for each sound.

Quality Differences. If the microphones have varying directivity with frequency, there are quality differences as well as intensity differences in the channels as the source moves. Angular localization is definitely affected by this. It has been found that the higher frequencies, where the head has relatively high directivity, contribute most to stereophonic localization. Localization tends toward the loudspeaker giving greatest high-frequency output, if the overall loudness is the same.

The very low frequencies contribute essentially nothing to stereophonic localization. For example, poor localization results if 1000-cycle low-pass filters are inserted, and no difference in localization is produced by eliminating frequencies below 300 cycles. It has been found[4,10,40] that much of the stereophonic effect is preserved if low frequencies are reproduced from only one low-frequency unit and side channels reproducing only frequencies above 300 cycles are employed. This is of great practical value for economical stereophonic reproduction such as home music systems. For the flexibility and high fidelity demanded by motion-picture and auditorium reproduction its use appears questionable until a great deal more study of it has been reported. The Philips tests[10] employed microphones a small distance apart; with widely spaced microphones characterizing the practice in this country serious pickup difficulties can be foreseen, as well as "crossover" complications in the loudspeaker systems. For "special effects loudspeakers," however, the low frequencies do not appear necessary if the main object is to obtain localization.

Arrival-Time Differences. Another phenomenon affecting angular localization is the change in arrival time of the direct-sound pulses from the several loudspeakers as the source moves upon the stage. These differences were mentioned above, and were shown to be considerably greater than those ordinarily encountered in simple binaural hearing. For example, in Fig. 3 the right and left channels reproduce sound pulses from the source later than the center channel by time intervals corresponding to distances *a* and *b*, respectively. The observer does not recognize the three pulses as distinct. However, it has been shown[17,41] that localization tends towards the loudspeaker which reproduces the earliest pulse. These effects have been called "Fusion" and the "Precedence Effect" by the authors of Ref. 17, who give a clear and detailed discussion of their relation to stereophonic reproduction. Qualitatively their discussion applies to stereophonic reproduction in general, but the precise data on precedence is limited to time differences of 2 msec or less, whereas common stereophonic conditions produce differences much greater than this. The following qualitative statements are deduced from this writer's own experience:

(a) The effect of arrival time is to make localization tend toward the loudspeaker from which the pulse arrives first.

(b) This effect is strong for small differences, say up to 3 or 4 msec, and tends to become weaker for greater time differences.

(c) The effect is relatively independent of where the differences are produced, whether on the pickup stage, in the listening room, or in the reproducing channels. Therefore differences in one section add to those in another, or can be made to compensate each other.

(d) These effects can be largely compensated by intensity or quality differences inserted in the channels, for any one observing position.

This effect acts to reinforce the

intensity effect for movement on the pickup stage. As a source moves toward a microphone the arrival time is advanced at the same time that intensity is increased. This is one of the important factors not duplicated by the bridged system. An interesting application is described by Grignon[39] in the triangular microphone arrangement for assuring center localization while maintaining stereophonic quality for a soloist or small source. Here the small advance of arrival time on the center microphone holds localization to the corresponding loudspeaker.

Reverberation. A fourth factor that might contribute to angular localization is ratio of direct to reverberant sound. Experience has shown, however, that it plays a very minor part in angular localization.

Dynamic Localization. Moir and Leslie[18] provide a very interesting observation on localization, as follows: "...dynamic localization of a source appears to be appreciably more accurate than is shown by the data obtained from localization tests on a stationary source. This applies to all variations of two- and three-channel systems that we have compared."

Depth Perception

Depth perception in stereophonic reproduction is controlled by essentially the same factors as in ordinary binaural listening described above, viz.: absolute intensity, quality, and ratio of direct to reverberant sound.[27] As the sound intensity decreases, the impression is produced of the sound moving away. The same illusion accompanies a relative loss of high frequencies. The most important contributor to the feeling of depth, however, is change in the ratio of direct to reverberant sound on the pickup stage. As the reverberant energy becomes more prominent, the source appears to recede on the virtual stage. In practice the microphones are closer to the sound sources than listeners would be, and changes in direct-to-reverberant sound ratio can be heightened to give more definite impressions of depth on a virtual stage than are created on a real stage. This can be seen in Fig. 1 of Ref. 27. As in ordinary listening, however, depth localization is less precise than angular localization.

Effect of Observer Position

Up to this point, for the sake of simplicity, the paper has been written as if all observing positions were equally good. Actually this is far from the case, as all experimenters have pointed out. From the standpoint of the practical use of stereophonic reproduction in the theater, this is a truly serious problem. Here the very factors which produce the stereophonic effect prove a disadvantage in some aspects, and measures must be taken to compensate them.

Source Position Shift as Observer Moves. The effects so far described characterize listening at the position of Fig. 3, or other listening positions on the center line where the distances to the side loudspeakers are equal. They also apply to other observing positions qualitatively, but as the observer moves away from the center large shifts of virtual source position may occur. The stereophonic feeling of spaciousness is preserved, and virtual sources continue to move, but they are not localized at the same place on the stage by all listeners as they would be on a real stage.

Figure 4 illustrates what is happening, for a source at center of the pickup stage, and a typical setup. Observer 1 receives identical direct sound pulses from the two side channels. Even here, however, the center-channel sound arrives slightly ahead of that from the sides, and at greater amplitude. In practice, the center channel is operated at lower gain than the side channels to correct for this.

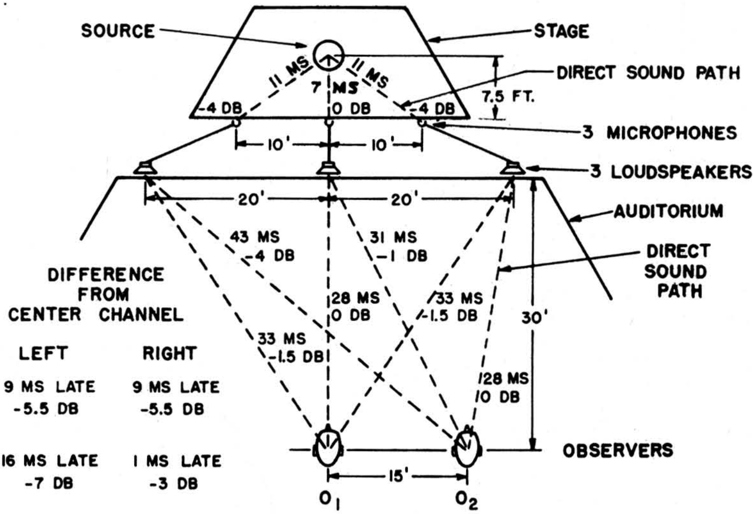


Fig. 4. Effect of changing listening location. As the observer moves away from the center-line of the auditorium the sound from the "near" loudspeaker increases in intensity and decreases in relative arrival time, making the virtual source shift in the same direction.

Observer 2 at the right receives pulses from the three loudspeakers with the relative times and intensity levels shown. It is seen that the righthand loudspeaker now contributes both a more intense signal and an earlier signal than before; and both of these effects are known to make localization tend in its direction. This is indeed the case, and as the observer moves to the right the virtual source position moves in the same direction. Note that the differences in time are several milliseconds. Qualitatively (again based upon personal experience) it is found that a considerable shift takes place for small observer deviations from the center, where relative intensity changes are small. These must be ascribed to changes in arrival time. For any given observer position these shifts can be compensated by changes in channel gains, and appear to become relatively constant at anything over a few milliseconds. Obviously the effects of intensity increase can be overcome by unbalancing the channel gains.

Methods of Reducing Shifts. The patent of Ref. 41 contains a suggested method of alleviating these troubles. The loudspeakers would be so designed as to project a delayed signal and one of reduced intensity in the forward direction compared to the side directions. This would tend to equalize conditions for the various observing positions.

Suppose that observer 2 remains at the right while the source moves to the left. The intensity increases in the left channel, but more important the arrival times become more nearly equalized, and the virtual source moves toward the left. Only the intensity change is duplicated in the bridged channel, so that there is definite advantage in the real system considering all observing positions. If the source moves to the right, the arrival time disparity is aggravated; but since there appears to be a limit to the effect of arrival time this negative effect is smaller than the positive advantage for movement to the left, and an overall gain results.

If the observer turns his head to follow movement of the virtual source, the effect is to oppose the movement, since the ear on the side of the head in the direction of movement in effect turns away from the loudspeaker of increasing intensity, and the opposite ear turns toward the loudspeaker of decreasing intensity. Since the sound tends to move "too fast" toward the microphone being approached because of the combined effects of intensity and arrival time, this is an advantageous compensating factor, considering all seats in the auditorium.

APPLICATION OF BASIC PRINCIPLES

The practical art of applying stereophonic reproduction for public use is now building up rapidly, and many papers may be expected in the future. The various references contain data on the small number of tests made previous to 1952, notably Ref. 39 in which Grignon describes tests specifically designed to determine techniques applicable to motion-picture production. The present paper is concerned primarily with the underlying principles, but it seems useful to give some illustrative examples of how they are used. These examples are primarily of situations with which the author has had personal experience.

Number of Channels

The number of channels will depend upon the size of the stage and listening rooms, and the precision in localization desired. Two channels give a large measure of the spacious effect desired for stereophonic reproduction, and will give fairly accurate localization for a small stage. Such a system on an ordinary-sized stage will give quite different localization impressions to observers in different parts of the auditorium, and is apt to suffer from the "hole-in-the-center" effect where all sounds at center stage seem to recede toward the back. Nevertheless, for a use such as rendition of music in the home, where economy is required and accurate placement of sources is not of great importance if the feeling of separation of sources is preserved, two-channel reproduction is of real importance.

That this is true is borne out by the current sponsored programs being broadcast by radio stations in various parts of the country using the FM transmitter for one channel and the AM transmitter for the other. Experience with this service in the writer's home has demonstrated the great increase in enjoyment it provides. Various methods for utilizing a single carrier for this type of broadcasting have been proposed,[18,40,42] using upper and lower sidebands separately, simultaneous AM and FM modulation, and modulating one channel on a subcarrier which is then modulated with the other channel on a regular FM transmitter. For such service the idea of supplying only one low-frequency loudspeaker appears important. It is well to recognize that a poor crosstalk ratio between channels in such a stereophonic system is not serious, because the relative intensity levels in the two channels never become greatly different. Thus systems which could not be considered for separate programs may be usable for stereophonic reproduction.

Three channels appear to be a good economic choice for ordinary stages and auditoriums. Good accuracy of localization can be achieved for favorable observing positions, with reasonable results at other seating locations. The center channel is a great aid for solo and close-up work, as well as removing the "hole-in-the-center" effect mentioned above. For unusually wide stages, additional channels have been found necessary.[43,44] At present it may be taken as a rule of thumb that additional channels should be considered when stage dimensions require channels spaced more than 25 ft on centers.

Loudspeakers

Placement. Loudspeaker placement is straightforward if considered for sound alone. The outside loudspeakers are placed at the outside edges of the space considered the reproducing stage, since sound cannot be made to travel past the outside speakers. The center, or other loudspeakers are placed at uniform spacing across the stage. It was stated that the close microphone position ordinarily used makes it possible to enhance depth effects. The source can therefore be made to appear in front of the loudspeakers, and they may be placed a few feet back of the front of the stage. In the Bell System demonstration at Carnegie Hall in 1940, the outside loudspeakers were spaced 40 ft on centers, and the front of each loudspeaker was 11 ft back of the decorative sound-transparent front curtain.[35] This curtain was illuminated in various simple color patterns during the performance, an artifice which adds enjoyment when no picture accompanies the sound.

For sound-picture reproduction, the effect of the picture is great, and the precision of localization required is smaller. If the sound tends to be in the region of the visible source, it will be localized there. Consequently here it *is* possible to create the illusion of sound outside the farthest loudspeaker.

When the stereophonic system is used for sound reinforcement serious difficulty may be experienced in placing the loudspeakers where they will not obstruct the view. Fortunately here, also, the source is visible. In addition, it was shown that localization in the vertical plane is poor. The loudspeakers can therefore be placed above or below the stage level without loss of illusion provided high fidelity of reproduction is maintained. It is also sometimes possible to use a smaller loudspeaker in the central positions, without full low-frequency response, to give proper localization. One of the most successful stereophonic reinforcement systems was tested in the Hollywood Bowl in 1936,[45] where the loudspeakers were mounted on a platform 45 ft above the stage level. The system supplied almost uniform sound level throughout the seating area, and considerable amplification even for the closest seats. Nevertheless the illusion that the sound came directly from the orchestra in the shell was excellent. To preserve a good illusion the loudspeakers should have approximately the same spacing as the channel microphones.

Characteristics. Since the illusion is caused by the receipt of multiple sound pulses, and in view of the observer-position effects discussed above, it is important that the loudspeakers give uniform angular coverage of the whole seating area. Actually, according to the disclosure of Ref. 41, greater energy should be supplied to seats at the side than to those in front of a loudspeaker, the inverse of the ordinary loudspeaker directional characteristic. Some toeing-in of the outside loudspeakers will help the average situation. In addition to these factors, de Boer[4] also recommends minimizing sound projection to areas outside the audience to reduce wall reflections, and maintaining the quality of the several channels above 300 cycles as alike as possible. Quality differences will be interpreted in the stereophonic illusion as differences in direction.

Bridged Loudspeakers. It is possible to bridge a center loudspeaker across the outside channels, which has the effect of reducing the apparent stage width.[6,27] This would be useful if it were impossible to place the side loudspeakers as close together as desired. It would be subject to the limitations of bridged systems already pointed out.

Microphones

Placement. Microphone placement may be simple or complicated, depending on the application. From what has been said, it will be evident that creating the stereophonic illusion is a compromise between favorable and unfavorable factors, and microphone placement and movement can be used to advantage in effecting this compromise. Since the illusion depends upon differences in intensity and arrival time at the microphones, and change in ratio of reverberant to direct sound, the microphones must be placed close enough to the sources to create these differences. This means that each microphone "covers" only part of the stage and will be closer than fixed microphones placed for single pickup. If pickup of action is necessary in a room where ordinary reverberation times obtain, the necessity of close pickup is apt to accentuate depth effect, and require a small stage area. Then dimensions are multiplied if a larger reproducing stage is used, and the speed of movement on the pickup stage must be slowed by an appropriate factor. Conversely, if the action demands a large stage, special microphone-handling techniques such as those described by Grignon[39] will probably be necessary. A good combination is a dead stage in which a set of the size that will accommodate the action can be constructed with the proper combination of "flats" to give a reflected sound content that will produce the desired depth illusion.

The motion-picture industry is rapidly developing the art of microphone movement for stereophonic recording where action and movement of camera are all-important. For other stereophonic pickup, such as music, radio plays or sound reinforcement, fixed microphone positions aided by some mixed-in special pickups will usually suffice. The regular microphones are deployed in front of the stage. If all action is at front stage, the outside microphones should be at the outside edges. However, to secure the illusion of action on a rectangular stage requires a greater stage width at the rear line than at the front (Fig. 6 in Ref. 27), and some compromise must be made; so the side microphones are usually placed somewhat inside the edges. This is particularly true of a two-channel system where a compromise between "hole-in-the-center" sound and well-spread sound must be effected. In this connection, a bridged center microphone is frequently used and does fill up the hole for center observing positions. However, it obtains this effect by adding sound to the side channels at advanced arrival time, thus aggravating the shift of the virtual source as the observer moves to the side of the auditorium.

After considerable experimentation, the microphones for the Philadelphia Orchestra recordings demonstrated by the Bell System in 1940 were suspended 10 ft above the stage and 5 ft inside the front row of musicians. The orchestra width was about 40 ft and the outside microphones were 28 ft apart. For small stages with actors, good results were obtained with a 12 ft square stage in a very dead room, using two microphones 9 ft apart and 5 ft from the front of the stage. In a rather reverberant medium-sized room a stage 15 ft wide by 6 ft deep, using three channels, with the microphones on 6-ft centers and 4 ft from the front line, proved satisfactory. In this case, note the shallow depth dictated by the reverberation in the room.

Directivity. Directive microphones can frequently be used to advantage. Since to produce an angular illusion it is necessary to generate intensity differences in the channels, a study of the geometry will show that greater movement is required at the rear of the pickup stage than at the front to produce a given angular impression. If the microphones are directive, greater intensity changes will occur as a source moves across the stage from the lobe of one microphone into that of another, and the rear line will be shortened. At the front line the directivity effect may be so great that the sound appears to recede between microphones. Experiment has shown that with moderate directivity, and by toeing in the lobes of the side microphones somewhat, an advantageous compromise between these two effects can be made and better overall coverage of a rectangular stage obtained.

This effect may be obtained with microphones of uniform directive properties, such as the cardioid types, or with the directivity only at high frequencies characteristic of a relatively large condenser or dynamic microphone at normal incidence. The latter will give accentuated directional effects with less change in overall loudness. Here directional effects are really quality changes. While in monophonic reproduction these quality changes would be objectionable, in stereophonic work the listener's fused impression consists of the contribution from several sources and the source is always in the direct lobe of one microphone. If the normal-incidence characteristic of the microphone is considered in overall system performance, the fidelity will remain high from all source positions.

The elimination of pickup from behind the microphones is a definite advantage in most cases. Obviously it eliminates noise. But it also eliminates part of the reverberation, and since most stages have more than the desired reverberation ratio for the physical depth, this is an advantage.

Reverberation. A pickup problem which has received little study as yet involves the adaptation of the reproduction to the listening room. The concept for reproduction in a theater or concert hall appears straightforward. To get good localization requires close pickup, and therefore the radiated sound approaches in quality the direct sound that would have been projected into the theater by a live (if gargantuan-voiced) performance. The theater then applies its own acoustical characteristics to the sound. In broadcasting and phonograph reproduction, however, listening is usually done in small, rather heavily damped rooms, and monophonic microphone techniques have been worked out to give a pleasing amount of reverberation from the pickup stage. Without doubt, some way will have to be found to produce a similar effect in stereophonic reproduction with the closer pickup required.

Bridged Microphones. Since channels are expensive and the complications grow with greater numbers, it is tempting to use bridged microphones to simplify the system. If this technique is used with restraint to gain additional realism in reproduction, it can be extremely useful. If it is used in the hope that it will be a cheap way of duplicating the performance of a more elaborate system, the results are bound to be disappointing. The tests reported in Fig. 1 of our original article[27] demonstrate this and are worth careful study. Discussion offered above explains why such techniques cannot be expected to duplicate real stereophonic channels.

An example of a useful application of the bridged microphone is its use to emphasize a small group of instruments in an orchestra, when the overall pickup is satisfactory in other respects. This was employed in the Hollywood Bowl demonstration[45] where one extra microphone was used continuously on the right channel, and others were employed during special parts of the performance. In monophonic systems multiple microphone pickup often leads to poor fidelity because of cancellation between the signals from the microphones in specific frequency regions. In stereophonic systems this effect is ameliorated because sound is fused from several sources.

When a solo instrument or voice is to be employed with an orchestra, separate pickup is very effective. The microphone should be arranged to pick up as little as possible of the orchestra, and the output should be mixed into the orchestra channels to give the localization desired. By far the best result will be obtained if the three-microphone triangular pickup described by Grignon[39] is used. The soloist will then be localized by substantially the whole audience at the desired location and the realism will be enhanced over a single microphone pickup.

Amplifiers

Amplifiers for use in stereophonic systems do not differ from those of monophonic systems except in number. The characteristics of the amplifiers in the various channels should be similar, and the gain should be stable so that no undesired level differences will occur. It is usually found desirable to have a ganged volume control which will adjust the overall level, and an individual control in each channel for balance or intentional unbalance settings. Similar provisions for quality-changing networks are desirable. If bridging systems are to be used proper networks and bridging amplifiers must be provided to insure that signals flow only in the desired directions, and inadvertent gain changes are not made during switching. It is also good practice to observe a poling convention throughout all channels, including the microphones and loudspeakers, although the channel spacings are so wide that only very low frequencies can be considered at other than random phase in one channel compared to another.

As a matter of economics, it is probably true that the added complication of stereophonic reproduction will be employed only for high-fidelity reproduction. Consequently the amplifier systems will require the same care in design and attention to detail that is required to secure high fidelity in monophonic systems.

APPLICATION TO RECORDING

The general principles of stereophonic sound apply to reproduction whether it is from recordings or from direct transmission by wire or radio. Recording has problems of flutter and maintenance of time differentials between channels peculiar to itself, and in general yields more severe technical problems in maintaining low noise and distortion. Yet it is certain that the great bulk of listening hours will be provided by recorded material. The effect of such distortions in stereophonic recording is therefore of great importance.

Distortion

The consensus of reported opinion in the literature is that stereophonic reproduction reduces the objectionableness of distortion and noise.[18] This unwieldy word is used because no test data are available to show whether the distortions become less detectable by the observer, or whether he is willing to overlook more distortion because of the increased pleasure of listening provided by stereophonic sound. Doubtless both reasons are true in part. The most outstanding example of the latter is the preference of observers for stereophonic sound, even though seriously degraded in frequency band.

Subtractive Type. It seems probable that distortions of a "subtractive" nature are actually less detectable. A dip in response of a single loudspeaker, or the equivalent caused by cancellation between two microphones on the same channel, will not be so noticeable if sound contributions from other channels not so distorted are being fused with the distorted signal.

Flutter. By similar reasoning, it seems probable that flutter will not be as noticeable on stereophonic reproduction. It is well established[46] that small frequency variations in the signal are turned into much larger amplitude modulations by the sharp resonances of the listening auditorium, and these are detected by the ear. Each channel will excite a different resonant pattern in the room. The fusion effect should therefore reduce the resultant modulation at the ear, with consequent reduction in flutter sensitivity.

Additive Type. It does not seem likely that the actual detectability of "additive" effects such as noise and distortion-product frequencies would be decreased by stereophonic reproduction, but their degrading effect does seem to be lessened. In monophonic reproduction any noise (distortion products are equivalent to noise) competes directly with the signal for attention whereas in stereophonic reproduction the directional illusion separates noises and program in space and allows the observer to concentrate more on desired sounds. Moir and Leslie[18] report a 12-db improvement in signal-to-noise ratio "due to the ears' steerable directivity pattern."

Channel Differences

Quality. For ideal results the quality of the various channels should be identical. Differences in quality will show up as differences from desired localization. On the other hand, a stereophonic effect will be preserved even with fairly large differences in quality. Consequently, in practical operation the attention now given to maintenance of uniform frequency response in high-fidelity monophonic systems will be adequate for the channels of stereophonic recording systems.

Level. The level difference between channels should be kept small, but the requirement does not seem inordinately difficult. A 2-db unbalance between the channels of a two-channel system — the most critical case — would shift the virtual source about 4 ft across a 45-ft stage.

Time. The requirement of time-identity of scanning position for the channels is considerably more stringent for true binaural than for stereophonic reproduction. Fifty microseconds difference would cause a 5° shift in binaural localization, corresponding to 0.7 mils misalignment for 15 ips track speed. However, for a two-channel stereophonic system a severe requirement might be 1 msec, corresponding to 15 mils misalignment for 15 ips track speed. This amount, equivalent to approximately 1-ft distance difference, would correspond to an actor moving from one side of a chair to the other, or to an auditor shifting from one seat to the next in the theater.

Dubbing

In the process of preparing a recording for release, a very important function is dubbing-in sound effects and music, or re-recording with altered quality or balance. In stereophonic recording there is the added requirement of proper position of the sound. When a single source must be given position, use is made of a bridged system and a "pan-pot." This is an arrangement of attenuators on a common control which will feed to each channel an intensity simulating the intensity it would have received if the original recording had been made with multiple microphones. The characteristic of the instrument built for the Auditory Perspective demonstrations of January 1934 is

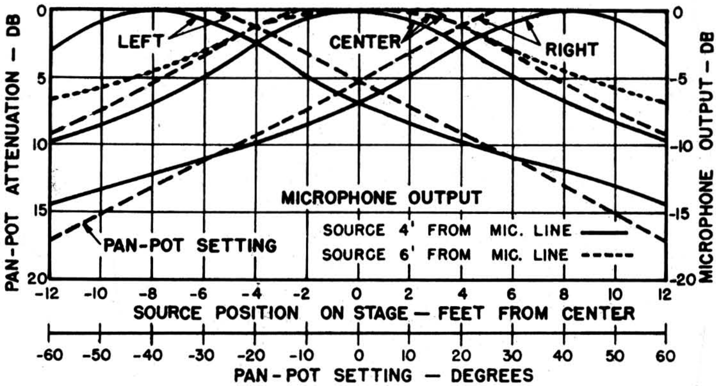


Fig. 5. Pan-pot characteristics. As a scource moves across a pickup stage, the direct sound microphone outputs vary as shown. The dashed line is the corresponding attenuation introduced by the pan-pot constructed for a 1934 demonstration.

shown by the dashed lines in Fig. 5. The control was made of three continuous-winding ladder volume controls modified with "bridges" for the sliders over parts of the angular range to give the flat portions of the curves. The solid lines of Fig. 5 show the variation in direct sound at each microphone for a source moving across a line 4 ft from the microphones, which are assumed 8 ft on centers.

It can be seen that this simple volume control scheme is a fair representation of the actual case. The dotted curve shows for comparison the variation in level for the center microphone when the source moves across a line 6 ft from the microphones. The difference between these curves emphasizes that the relationships vary for different stage depths, and in using a pan-pot the operator must adjust his settings to the desired effect. The curves also show the rather small level differences that exist. It will be seen that the pan-pot characteristic gives lower channel levels at "side" settings than the actual pickup. This is desirable to compensate for the absence of arrival-time and microphone directivity effects.

Disk Recording

The adaptability of tape- and film-recording methods to stereophonic sound is readily apparent, and these strip media are relatively unlimited as to number of channels. For two-channel recording, disk methods are also practicable. Two systems have been demonstrated. In one[47] two grooves are used in parallel, one starting near the outer edge and one near the middle of the recording area. Two reproducers are used. In the other[18,48,49] a single groove is used, with one channel recorded as a vertical and the other as a simultaneous lateral track. While the interaction or crosstalk between channels is relatively high, experiment has shown that a sufficient ratio for stereophonic work can be obtained.

CONCLUSION

Although stereophony is attaining a respectable age, much more information must be obtained before it can be said to rest on a foundation of quantitative relationships. It is hoped that this summary of present knowledge will stimulate the acquisition of this information, and in the interim will serve as a useful guide to those who must make recordings without waiting for complete theoretical understanding.

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Discussion

John G. Frayne (Westrex Corp.): Would the speaker tell us whether there's ever any possibility of duplicating the real stereophonic effect by using the artificial method of taking a monaural track and making it into stereophonic by manipulation of gain and equalization.

Mr. Snow: I don't think so, because by manipulation of the channels you do not duplicate all the effects which you can get on a real stage. As the speaker, let's say, walks across the stage you can get the actual effect of the intensity increase, you automatically get the effect of the arrival time with sound coming earlier from the nearest channel. You can use microphone directivity, if you use it with care, to enhance both of those effects; and it seems to me that, at least without something that I can't quite imagine in elaboration, it would be awfully hard for any one person to duplicate all these effects as he tried to twist some knobs. And, of course, there's another thing: no matter what you try to do in this way, you can do it for only one source at a time, if you're doing it artificially, whereas the actual pickup will handle any number of sources all at one time. My own feeling is that it is very unlikely that the completely artificial manipulation of channels will give you a real duplication of multiple-channel pickup at the original scene.

Dr. Frayne: In that case, would you say then that the industry is missing an opportunity of improved sound presentation by placing so much emphasis on the pan-pot method of producing stereophonic sound?

Mr. Snow: I would say that they ought to consider that something to get rid of as soon possible. It's something which can be used to advantage, I'm sure, in many situations; but I feel that it should be used only as a last resort, rather than as a first resort. It won't sound as good as the real pickup or the original.

Dr. Frayne: On the matter of the number of channels, I notice you say that three channels give a very good stereophonic effect. Now, in Cinerama, I believe, they use five stereophonic channels behind the screen and I am told by Cinerama engineers that they find a much better stereophonic effect by using five rather than three.

Mr. Snow: I use three for two reasons. One is that my personal experience has been with two or three and it makes the fundamentals easier to show. The fundamental principles I don't think would change with the number of channels, but I do feel that the number of channels depends upon the width of the stage, the width of the scene that you're going to cover and perhaps as a rule of thumb, you might say that a channel should not cover more than a width of 20 or 25 ft with a single channel. The cinerama screen is so much bigger than the 50-ft wide total that they needed more channels.

Dr. Frayne: In the case of CinemaScope, which uses in some cases a 65-ft screen, is it possible to cover that with three speakers?

Mr. Snow: I imagine that probably it will be thought so. I don't mean to sound as facetious as that. Actually, when you have a picture, you don't need to have as faithful sound localization as when you're only trying to reproduce an orchestra with nothing to look at, as I have usually done in my work. The picture certainly can complement the "monophonic" sound to some extent, as we're all well aware, since we've been getting along with one channel on any width screen up to now. As a matter of fact, I would think in a picture, up to the width that you spoke of, that would be satisfactory. I have no doubt, however, that more channels would be even more realistic, but it's certainly a matter of economics.

Dr. Frayne: What do you think of the auditorium speakers as adding to stereophonic effect?

Mr. Snow: That's something for the industry to decide now. I haven't had any personal experience with that. I have nothing against it. My feelings on stereophonic effects are that you manipulate the channels to get the effects you want. I was trying to point out the fundamentals that you have to preserve to get those effects, but I feel that when you get to the point of having auditorium speakers, and so on, it gets a little bit more in the showmanship angle than straight physics. And I'll leave that to the showman.

Edward S. Seeley (Altec Service Corp.): Do you believe it possible to recreate a location outside of the outermost speaker in a three-channel system?

Mr. Snow: Not acoustically, but with a picture I do think you can. However, there doesn't seem to be anything in the physics or physiology that I know of that would pull the sound past the outside loudspeaker just from the standpoint of localizing the sound with your eyes shut, but obviously if you have a picture, with a sound source that's outside the outside loudspeaker it's not very hard to imagine that the sound is pulled somewhat outside of the actual physical source of it. But you can see, from the standpoint of sound alone, that if you turned off all the channels but the one on the side that we're talking about, everybody would localize the sound right in that loudspeaker and there's nothing I can see that would make you pull it any further than that when the other ones are running.

Loren L. Ryder (Paramount Pictures Corp.): With respect to the remarks I am about to make, I will first say that my comments are not against stereophonic sound. Now with respect to what can be done by panning sound, we at Paramount have found that following some of the principles that were explained here but using phase displacement, rather than volume, we can more definitely control the placement of sound than by the volume difference between loudspeakers. We also find that equalization, as mentioned by the speaker, is a very strong control. We at Paramount have used displacement (phase shift) by as much as four and as high as seven sprocket holes in the control of sound placement. Having once established that type of sound placement, it makes little difference what volume is used from the three loudspeakers as far as the listener is concerned, and as far as his selection of a point source. Therefore, with such an arrangement, we can gain a proper directivity much further to the side of the theater and further down toward a side loudspeaker, than we have ever been able to obtain either by volume control or by classical stereophonic sound.

Mr. Snow: I'm very glad to hear of some practical experience along that line, because I certainly would expect that on the basis of the principles I was enunciating here; but unfortunately I have never been able to try it. Thanks very much for that comment.

Mr. Seeley: May I ask Mr. Ryder if his remarks apply to simultaneous sounds from distributed sources as well as to dialogue?

Mr. Ryder: My remarks apply to dialogue, music and sound effects. In the picture *Shane* there are sequences in which the violin section of the music is on the left-hand loudspeaker, the music base is on the right, dialogue is center screen, calls are heard from the left side of the screen and sound effects are moving back and forth.

We find no trouble in gaining proper placement of sound effects and we find no confusion when these sounds are ultimately reproduced in the theater. It seems to me that there is a great deal still to be learned in regard to the effective handling of sound when reproduced from three or more loudspeaker systems. For those who have not experimented with phase shifting, I recommend that they do so.

It is our feeling that there are a number of ways of gaining the same effectiveness to the audience. The real question is — which way is the simplest, least costly, and least subject to error and disturbing effects.

Richard H. Ranger (Rangertone, Inc.): I think that we all owe a debt of gratitude to Dr. Snow for this elucidation of these principles and I'd like to check again on what Mr. Ryder has just said, that timing has a terrific effect on directivity. We are indebted to Dr. Haas of Göttingen for work on this timing business, because he has elucidated this matter very intensively and confirms what has just been said. In other words, timing is of the utmost importance and you can actually get a curve or a correspondence, shall I say, between timing and intensity. In other words, as Mr. Ryder has just suggested here, you can move a subject across the stage just by timing; and you can also move it just by intensity. And you can do the corresponding thing of making them compensate each other. In other words, you can move the timing so as to make the apparent location move to the left, we'll say, and you can increase the intensity to hold it where it was. And you soon find, however, that when you do that the timing completely outweighs the intensity, so that actually the timing becomes in many ways the controlling factor. As to flutter and other quality factors, it has been my finding that they are entirely determined by the sound that you get first, or should I say that they are greatly determined by that. You can have considerable flutter, if you please, in the sound that comes later, and it will not affect the apparent quality at all. Timing and intensity, then, are terrifically important in these things. I don't quite go along with the statement that timing is the only essential, however. Perhaps Mr. Ryder did not intend to give that impression.

Mr. Ryder: It is certainly possible to control placement with intensity.

Col. Ranger: In fact, I feel that you can overdo the timing business, because you get a little bit of an uncertainty, if I might put it that way, when you get too much intensity from the wrong speaker, which you can do. You get a confusion of sound, so I feel that the answer is going to be a judicious use of the two to come up with the best quality.

Mr. Ryder: A further comment along the line of Col. Ranger's thoughts: if you use timing and equalization and volume, you have a very smooth complete control, and it's not as awkward to do as one would think. In this regard, we can refer to the picture *Shane*, which is largely handled by timing and not by volume. I should also comment that in all the work with respect to motion pictures where it is necessary to do much editing and cutting of motion pictures so that there is a change in sound placement on cuts, I personally favor a minimum movement of dialogue and a maximum use of stereophonic for punctuation in storytelling for effects and for music.

Walter Brecher (*Leo Brecher Theatres*): In connection with the finding that the number of channels to use with a wide screen should be based on a spacing of about 20 ft by channel, there are a great many theaters whose total width is in the neighborhood of 30 to 40 ft. It's my impression that there is a radius of illusion of approximately 15 ft which is centered on each speaker and in view of the acknowledgment that the visual pull does affect the illusion of location of sound source, does stereoscopic sound offer any substantial benefit for a theater of the dimensions that I have described?

Mr. Snow: I didn't mean to imply that. Let's put it another way. I meant that I felt that until people have actual data on it, that that was a fairly good rule of thumb as to the width where you might begin to consider that you might need more channels. But the stereophonic system will improve the reproduction in a living room where the loudspeakers are 5 ft apart or 12 ft apart, so that what I gave is in my opinion, a maximum width, and for anything smaller than that you can definitely get an improvement by using multiple channels. You might say, well, why not just use two channels? I believe that that would just make it more difficult from the pickup standpoint to get the effects you want, particularly when so much of the sound should come from the center of the stage for close-ups. When you have a third channel you can pretty nearly guarantee that for most of the seats in the auditorium. You're trying to build the illusion. With loudspeakers just at the sides, that's much harder to do.